Listing of Claims:

1. (Original) 1. A method of evaluating the processing delay of a speech signal contained in data packets received in a receiver terminal equipped with a telephony module during a voice call to a terminal sending said data packets over a packet-switched network, said method including the following steps:

obtaining from the received data packets a stream of audio packets containing the speech signal;

within a predetermined decoding time, decoding the stream of audio packets obtained and creating a first reconstituted speech signal;

duplicating at least a portion of the speech signal reconstituted by the telephony module constituting a second speech signal;

determining the delay difference between the first speech signal and the second speech signal; and

calculating the processing delay D3 of the speech signal in the receiver terminal from at least the measured delay difference between said first speech signal and said second speech signal and the predetermined decoding time.

2. (Original) A method according to claim 1, wherein the measured delay difference between said first speech signal and said second speech signal is measured by intercorrelation of the envelope signals of said first and second signals.

3. (Previously presented) A method according to claim 1, wherein the step of determining the delay difference is preceded by a step of detecting vocal activity in the first and second voice signals, the subsequent steps being executed if the vocal activity detected in the first and second signals is above a predetermined threshold.

4. (Previously presented) A method according to claim 1, wherein the step of decoding within a predetermined decoding time uses a decoding algorithm identical to that used in said telephony module or the decoding time difference whereof relative to the algorithm used in the telephony module is constant and known.

5. (Previously presented) A method according to claim 1, wherein the processing delay D3 is obtained by summing the determined delay difference between the first and second speech signals and the predetermined decoding time of the first speech signal.

6. (Previously presented) A method according to claim 1, wherein said packet switching network is an IP network and the data packets received in the terminal are IP packets.

7. (Previously presented) A method of evaluating the end-to-end transmission delay of a speech signal received in a receiver terminal during a voice call to a terminal sending said speech signal over a packet-switched network, the method including a step of evaluating the processing delay D3 of the speech signal in the receiver terminal by a method according to claim 1.

8. (Original) A method according to claim 7, further including the following steps:

evaluating the send processing delay D1 of the speech signal;

measuring the transmission delay D2 of the speech signal in the network; and

evaluating the end-to-end transmission delay from said send processing delay D1, said transmission delay D2 and said receive processing delay D3.

9. (Original) A method according to claim 8, wherein the send processing delay D1 of the speech signal is evaluated by consulting a table stored in the receiver terminal containing a predefined maximum value and a predefined minimum value of said delay D1 for each type of speech signal send coder, said predefined values taking into account the payload of the IP packets received.

10. (Previously presented) A method according to claim 8, wherein the transmission delay D2 of the speech signal in the network is evaluated using a Ping technique.

11. (Previously presented) A method according to claim 8, wherein the transmission delay D2 of the speech signal in the network is evaluated from sender report information extracted from the packets received.

12. (Previously presented) A method according to claim 7, wherein the end-to-end transmission delay is evaluated by summing said send processing delay D1, said transmission delay D2 and said receive processing delay D3.

13. (Previously presented) A method according to claim 7, further including the steps of:

creating information representing the end-to-end delay values obtained; and

sending said end-to-end delay information over the network to a collection server adapted to manage end-to-end delay information sent by a set of communication terminals connected to the network.

14. (Original) A device adapted to be installed in a receiver terminal equipped with a telephony module to evaluate the processing delay of a speech signal from data packets received in the receiver terminal during a voice call to a terminal sending said data packets over a packet-switched network, said device including:

a network filter module adapted to obtain a stream of audio packets containing the speech signal from the data packets received;

a control decoder module having a predetermined decoding time for decoding the stream of audio packets obtained and creating a first reconstituted speech signal;

an audio filter module adapted to duplicate at least a portion of the speech signal reconstituted by the telephony module, said portion of the reconstituted speech signal constituting a second speech signal;

means for determining the delay difference between the first speech signal and the second speech signal; and

means for calculating the processing delay D3 of the speech signal in the receiver terminal from at least the measured delay difference between said first speech signal and said second speech signal and the predetermined decoding time.

15. (Currently amended) A device according to claim 14, ~~adapted to be installed in a receiver terminal equipped with a telephony module to evaluate the processing delay of a speech signal from data packets received in the receiver terminal during a voice call to a terminal sending said data packets over a packet-switched network, said device including:~~

~~a network filter module adapted to obtain a stream of audio packets containing the speech signal from the data packets received;~~

~~a control decoder module having a predetermined decoding time for decoding the stream of audio packets obtained and creating a first reconstituted speech signal;~~

~~an audio filter module adapted to duplicate at least a portion of the speech signal reconstituted by the telephony module, said portion of the reconstituted speech signal constituting a second speech signal;~~

~~means for determining the~~ wherein the delay difference between the first speech signal and the second speech signal is measured by intercorrelation of the envelope signals of first and second speech signals;

~~means for calculating the processing delay D3 of the speech signal in the receiver terminal from at least the measured delay difference between said first speech signal and said second speech signal and the predetermined decoding time; and~~

~~means for implementing a method of evaluating the processing delay of a speech signal as claimed in claim 2~~.

16. (Previously presented) A device for evaluating the end-to-end transmission delay of a speech signal, adapted to be installed in a receiver terminal equipped with a telephony module to evaluate said transmission delay from data packets received in the receiver terminal during a voice call to a terminal sending said data packets over a packet-switched network, said device comprising means for implementing a method of evaluating the end-to-end transmission delay as claimed in claim 7.

17. (Previously presented) Telephone terminal equipment on a packet-switched network, in particular an IP telephone or a personal computer equipped with telephony software, said telephone terminal equipment including a device for evaluating the processing delay of a speech signal as claimed in claim 14.

18. (Original) Telephone terminal equipment on a packet-switched network, in particular an IP telephone or a personal computer equipped with telephony software, said telephone terminal equipment including a device for evaluating the end-to-end transmission delay of a speech signal as claimed in claim 16.

19. (Previously presented) A computer program on an information medium, including program instructions adapted to execute a method according to claim 1 if said program is loaded into and executed in an electronic data processing system.

20. (Previously presented) Computer program on an information medium, including program instructions adapted to execute a method according to claim 7, if said program is loaded into and executed in an electronic data processing system.

21. (New) A device according to claim 14, further comprising means for detecting vocal activity in the first and second speech signals, the delay difference between the first and second speech signals being determined if the vocal activity detected is above a predetermined threshold.